Grandstream Networks, Inc.

HT70X Analog Telephone Adaptor
HT701 HT702 HT704

HT701

HT702

HT704

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WELCOME

Thank you for purchasing Grandstream's HT70X, the affordable, feature rich Analog Telephone Adaptor. Grandstream HandyTone70X is a new addition to the popular HandyTone ATA product family. It features the rich audio quality, a broad range of voice codecs, and functionality including one independent SIP account per FXS port.

This manual will help you learn how to operate and manage your HandyTone70X Analog Telephone Adaptor and make the best use of its many upgraded features including simple and quick installation, 3-way conferencing, direct IP-IP Calling, and new provisioning support among other features. This HT70X is very easy to manage and configure, and is specifically designed to be an easy to use and affordable VoIP solution for both the residential user and the teleworker.

Safety Compliances

The HT70X phone complies with FCC/CE and various safety standards. The HT70X power adaptor is compliant with UL standard. Only use the universal power adapter provided with the HT70X package. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

Warranty

If you purchased your HT70X from a reseller, please contact the company where you purchased your device for replacement, repair or refund. If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number before you return the product. Grandstream reserves the right to remedy warranty policy without prior notification.

Caution: *Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty. Please do not use a different power adaptor with the HT70X as it may cause damage to the products and void the manufacturer warranty.*

- This document contains links to HT70X GUI Interfaces. Please download these examples for your reference here: http://www.grandstream.com/products/ht_series/ht701/documents/ht70x_gui.zip
- This document is subject to change without notice. The latest electronic version of this user manual is available for download at: http://www.grandstream.com/products/ht_series/ht701/documents/ht70x_usermanual_english.pdf

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CONNECT YOUR HT70X

Equipment Packaging

The HT70X ATA package contains:

- One HT70X Main Case
- One Universal Power Adaptor
- One Ethernet Cable
- One Vertical Stand (Only on HT702 and HT704 Packages)

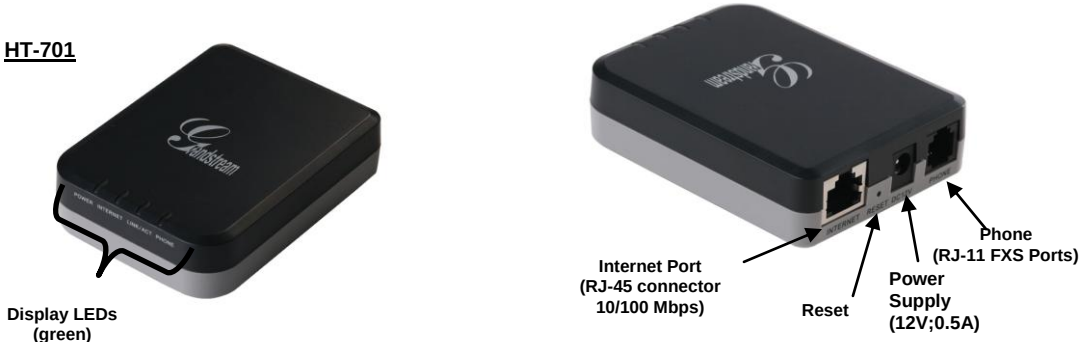
Connecting the HT70X

The HT70X is designed for easy configuration and easy installation. Configure the HT70X following the directions in the Configuration section of this manual.

1. Insert a standard RJ11 telephone cable into the Phone port and connect the other end of the telephone cable to a standard touch-tone analog telephone.
2. Insert the Ethernet cable into the Internet or LAN port of the HT70X and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)
3. Insert the power adapter into the HT70X and connect it to a wall outlet.

The HT70X Analog Telephone Adaptor is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services. The HT70X VoIP features and functions are available using a regular analog telephone.

Figure 1: Connecting the HT70X



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HT702 **HT704**

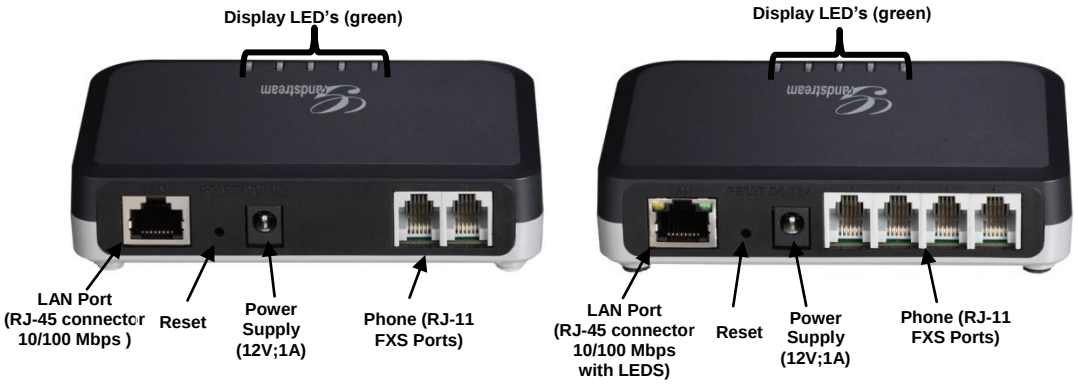

Table 1: Definitions of the HT70X Connectors

DC 12V	Power adapter connection
Internet Port (RJ-45)	Connect to the internal LAN network. **(HT701 Only)**
LAN Port (RJ-45)	Connect to the internal LAN network. **(HT702 and HT704 Only)**
RESET	Factory Reset button: Press for 7 seconds to reset factory default settings.
Phone Port(s) (RJ-11)	FXS port: to be connected to analog phones / fax machines.

There are four (4) LED buttons that help you manage the status of your HandyTone **701**, and there are five (5) LED buttons that help you manage the status of your HandyTone **702** and **704.**

Table 2: Basic Definitions of the HT70X LEDs Pattern

LEDs	
POWER LED	Indicates Power. Remains ON when power is connected
Internet LED	Indicates Access to Internet. Remains ON while there is Access **(HT701 and HT702 Only)**
Link/Activity LED	Indicates if There is Activity on the Internet Port **(HT701 and HT702 Only)**
PHONE LED	Indicate status of the respective FXS Ports-PHONE on the back panel Unregistered – OFF Registered and Available – ON (Solid Green) Off-Hook / Busy – Blinking every second Slow blinking FXS LEDs indicates voicemail

NOTE: All LEDs display green when ON

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Table 3: Advanced Definitions of the HT70X LEDs Pattern

Patten Number	Condition	LED	LED Behavior
LED-01	Device has normal power	Power	ON
LED-02	Power Error: Power is removed from the device or power supply with improper voltage is plugged in	Power	OFF
LED-03	Line X is registered normally to the sip providers network and is ready to make a call	Phone	ON
LED-04	Voice mail waiting for Line X	Phone	1sec ON / 3sec OFF
LED-05	Device has normal WAN connection and has obtained IP address	Internet	ON
LED-06	Internet link error: Device is powered up and ready to connect to the Internet but the WAN/INTERNET port is down	Internet	OFF
LED-07	Internet DHCP Error: Device is properly connected but it is unable to retrieve an IP address from the device it is connected to	Internet	0.25sec ON/ 0.25sec OFF
LED-08	Line Registration failed: Device is properly setup, can connect to provider's network, but cannot register to provider's SIP proxy (no 200 OK)	Phone	OFF
LED-09	Device is connected (has physical data link) but there are incorrect network settings typically associated with PPPoE connection failure	Internet	0.25sec ON/ 0.25sec OFF
LED-10	Hazardous potential test failed: Hazardous AC or DC voltage is present on the tip and ring or both signals of phone line X	Phone	0.25sec ON/ 0.25sec OFF
LED-11	Foreign electro Motive Force (EMF) Test fail. Foreign voltage is present on the tip, ring or both signals of phone line. Device has detected additional external Phone voltage on the FXS phone line.	Phone	0.25sec ON/ 0.25sec OFF
LED-12	Resistive fault test failed. Either tip or ring is shorted to ground or they are shorted to each other.	Phone	0.25sec ON/ 0.25sec OFF
LED-13	Receiver off hook test fail. One or more phones are off hook on phone line during test.	Phone	0.25sec ON/ 0.25sec OFF
LED-14	REN test failed – high REN detected. Too many parallel phones connected to phone line X	Phone	0.25sec ON/ 0.25sec OFF
LED-15	Line is active	Phone	1 sec ON/ 1 sec OFF
LED-16	Line inactive	Phone	ON
LED-17	During Provisioning Stage*	Internet / Phone	0.2sec ON/ 0.2sec OFF
LED-18	During Firmware Recovery Stage*	Internet / Phone	0.4sec ON/ 0.4sec OFF

***Note:** In Provisioning and Firmware Recovery Stage, the power LED is Steady ON.

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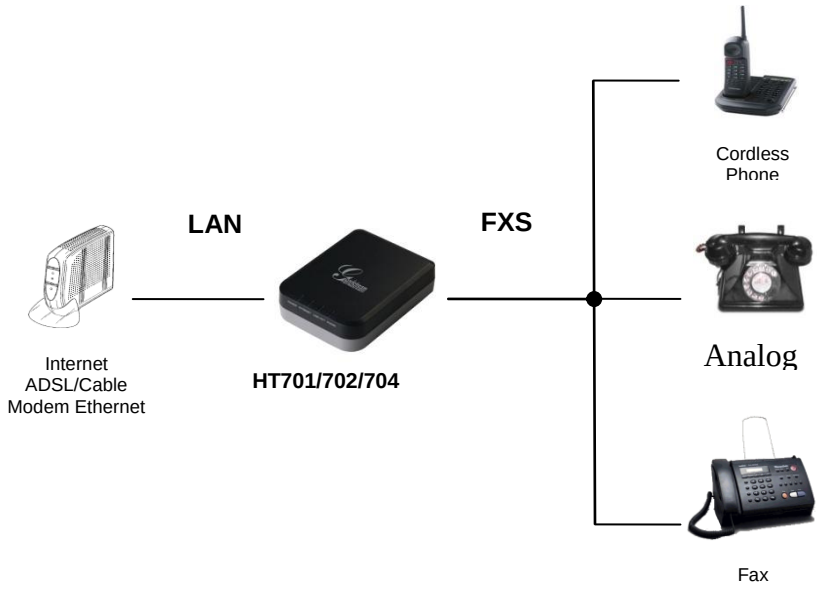
Figure 2: HT70X Connection Diagram



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PRODUCT OVERVIEW

The HT70X is a full feature voice and fax-over IP device that offers a high-level of integration including a 10M/100Mbps network port and one FXS telephone port, market-leading sound quality, rich functionalities, and a compact and lightweight design. The VoIP network signaling protocol supported is SIP. The HT70X is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market. Moreover, it supports comprehensive voice codecs including G.711 (a/μ-law), G.723.1, G.726-32, G.729 and iLBC.

Software Features Overview

- Supports Voice Codecs:
- G.711 (a/μ-law), G.723.1, G.726-32, G.729 and iLBC.
- T.38 Fax
- Comprehensive Dial Plan support for Outgoing calls.
- G.168 Echo Cancellation
- Voice Activation Detection (VAD), Comfort Noise Generation (CNG), and Packet Loss Concealment (PLC)
- Supports PSTN/PBX analog telephone sets

Table 4: HT70X Software Features

	HT 701	HT 702	HT 704
Telephone Interfaces	1 FXS ports	2 FXS ports	4 FXS ports
SIP Provisioning	1 Sip Account, 1 Profile	2 Sip Accounts, 2 Profiles	4 Sip Accounts, 2 Profiles
Number of Concurrent Calls	1 Concurrent Call	2 Concurrent Calls	4 Concurrent Calls
Voice over Packet Capabilities	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), Dynamic Jitter Buffer, Modem detection & auto-switch to G.711, Packetized Voice Protocol Unit (supports RTP/RTCP protocol), G.168 compliant Echo Cancellation, LEC (line echo cancellation) with NLP, Asymmetric RTP stream		
Voice Compression	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.723.1, G.726-32(ADPCM), G.729, iLBC, G.726-32 provides proprietary VAD, CNG, and signal power estimation, Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet header		
DHCP Server/Client	Yes, DHCP Client only		
Telnet Server	Yes		
Fax over Ip	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through, Fax Datapump V.17, V.19, V.27ter, V.29 for T.38 fax relay		
QoS	Diffserve, TOS, 802.1 P/Q VLAN tagging		
Transport Protocol	RTP/RTCP		
DTMF Method	Flexible DTMF transmission method, user interface of In-audio, RFC2833, and/or SIP Info		
IP Signaling	SIP (RFC 3261)		
Provisioning	TFTP, HTTP, HTTPS, TR-069, XML		

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Control	TLS/SIPS , SIP over TCP/TLS
Device Management	Web interface or via secure encrypted AES or non-encrypted central configuration file for mass deployment using Grandstream binary file or xml format. Auto/manual provisioning system or via built-in IVR. NAT-friendly remote software upgrade (via TFTP/HTTP/HTTPS) for deployed devices including behind firewall/NAT. Syslog support. Full support of TR-069 management protocol and Telnet access.
Dial Plan	*Yes*
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, and DTMF-based CID
Call Handling Features	Caller ID display or block, Call waiting caller ID, Call waiting/flash, Call transfer, hold, forward, 3-way conferencing, message waiting, Do-Not-Disturb (DND), call-return service

Hardware Specification

The table below lists the Hardware and Technical specification of HT70X.

Table 5: HT70X Hardware and Technical Specifications

	HT701	**HT702**	**HT704**
Telephone Interfaces	1 RJ11 FXS port	2 RJ11 FXS ports	4 RJ11 FXS ports
Network Interface	1 RJ45 10/100 Mb Base-TX, Full Duplex		1 RJ45 10/100 Mb Base-TX, Full Duplex port **with connectivity LEDs**
LED Indicators	POWER, INTERNET, LINK/ACTIVITY, PHONE	POWER, INTERNET, LINK/ACTIVITY, PHONE1, PHONE2	POWER, PHONE1, PHONE2, PHONE3, PHONE4
Factory Reset Button	Yes		
Universal Switching Power Adaptor	*Input:* 100–240 VAC/50-60 Hz 0.18A Max *Output:* 12VDC, 0.5A, **UL** certified		*Input:* 100–240 VAC/50-60 Hz 0.18A Max *Output:* 12VDC, 1A, **UL** certified
Environmental	Operational: 32°–104°F or 0°–40°C Storage: 14°–140° F or -10 ° –60° Humidity: 10–90% Non-condensing		
Dimensions (H x W x D)	26 x 65 x 86mm	28 x 115 x 75mm	28 x 115 x 75mm
Short Haul Loop	5REN, Up to 1Km on 24 AWG wire	3REN: Up to 1Km on 24 AWG line	
Polarity Reversal / Wink	Yes		
EMC	FCC part15 Class B, EN55022, EN55024, CISPR22, and CISPR24		
Safety	EN60950-1 & UL60950-1 (UL only for PSU)		
Compliance			

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BASIC OPERATIONS

Understanding HT70X Voice Prompt

HT70X has a built-in voice prompt menu for simple device configuration. The IVR menu and the LED button work with any of the FXS port. Pick up the handset and dial "*******" to use the IVR menu.

Table 6: HT70X IVR Menu Definitions

Menu	Voice Prompt	Options
Main Menu	"Enter a Menu Option"	Press "*****" for the next menu option Press "#" to return to the main menu Enter 01-05, 07,10, 13-17,47 or 99 menu options
01	"DHCP Mode", "Static IP Mode"	Press "**9**" to toggle the selection If using "*Static IP Mode*", configure the IP address information using menus 02 to 05. If using "*Dynamic IP Mode*", all IP address information comes from the DHCP server automatically after reboot.
02	"IP Address " + IP address	The current WAN IP address is announced If using "*Static IP Mode*", enter 12 digit new IP address. **You need to reset the HT for the new IP address to take Effect.**
03	"Subnet " + IP address	Same as menu 02
04	"Gateway " + IP address	Same as menu 02
05	"DNS Server " + IP address	Same as menu 02
07	Preferred Vocoder	Press "**9**" to move to the next selection in the list: • PCM U / PCM A • iLBC • G-726 • G-723 • G-729
10	"MAC Address"	Announces the Mac address of the unit.
13	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
14	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
15	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Press "**9**" to toggle between **TFTP / HTTP / HTTPS**
16	Firmware Version	Firmware version information.
17	Firmware Upgrade	Firmware upgrade mode. Press "**9**" to toggle among the following three options: - always check - check when pre/suffix changes - never upgrade
47	"Direct IP Calling"	Enter the target IP address to make a direct IP call, after dial tone. (See "*Make a Direct IP Call*".)
86	Voice Mail	Number of Voice Mails
99	"RESET"	Press "**9**" to reboot the device Enter MAC address to restore factory default setting (See **Restore Factory Default Setting** section)
	"Invalid Entry"	Automatically returns to main menu

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	"Device not registered"	This prompt will be played immediately after off hook If the device is not register and the option "Outgoing Call without Registration" is in NO

Five Success Tips when using the Voice Prompt

1. "*" shifts down to the next menu option
2. "#" returns to the main menu
3. "9" functions as the ENTER key in many cases to confirm or toggle an option
4. All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. For IP address, add **0** before the digits if the digits are less than 3 (i.e. - 192.168.0.26 should be key in like 192168**00**0**0**26. No decimal is needed).
5. Key entry cannot be deleted but the phone may prompt error once it is detected

Placing a Phone Call

Phone or Extension Numbers

1. Dial the number directly and wait for 4 seconds (Default "No Key Entry Timeout"); or
2. Dial the number directly and press # (Use # as dial key" must be configured in web configuration).

Examples:

1. Dial an extension directly on the same proxy, (e.g. 1008), and then press the # or wait for 4 seconds.
2. Dial an outside number (e.g. (626) 666-7890), first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 4 seconds. Check with your VoIP service provider for further details on prefix numbers.

Direct IP Calls

Direct IP calling allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy.

Elements necessary to completing a Direct IP Call:

1. Both HT70X and other VoIP Device, have public IP addresses, or
2. Both HT70X and other VoIP Device are on the same LAN using private IP addresses, or
3. Both HT70X and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

HT70X supports two ways to make Direct IP Calling:

Using IVR

1. Pick up the analog phone then access the voice menu prompt by dial "***"
2. Dial "**47**" to access the direct IP call menu
3. Enter the IP address after the dial tone and voice prompt "Direct IP Calling"

Using Star Code

1. Pick up the analog phone then dial "***47**"
2. Enter the target IP address.

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Note: **NO** dial tone will be played between step 1 and 2.

Destination ports can be specified using "*****" (encoding for ":") followed by the port number.

Examples of Direct IP Calls:

a) If the target IP address is 192.168.0.160, the dialing convention is
***47 or Voice Prompt with option 47, then 192*168*0*160.**
followed by pressing the "**#**" key if it is configured as a send key or wait 4 seconds. In this case, the default destination port 5060 is used if no port is specified.

b) If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:
***47 or Voice Prompt with option 47, then 192*168*0*160*5062** followed by pressing the "**#**" key if it is configured as a send key or wait for 4 seconds.

NOTE: When completing direct IP call, the "**Use Random Port**" should set to "NO". You cannot make direct IP calls between FXS1 to FXS2 since they are using same IP.

Call Hold

Place a call on hold by pressing the "flash" button on the analog phone (if the phone has that button). Press the "flash" button again to release the previously held Caller and resume conversation. If no "flash" button is available, use "hook flash" (toggle on-off hook quickly). You may drop a call using hook flash.

Call Waiting

Call waiting tone (3 short beeps) indicates an incoming call, if the call waiting feature is enabled. Toggle between incoming call and current call by pressing the "flash" button. First call is placed on hold. Press the "flash" button to toggle between two active calls.

Call Transfer

Blind Transfer

Assume that call Caller A and B are in conversation. A wants to *Blind Transfer* B to C:

3. Caller A presses **FLASH** on the analog phone to hear the dial tone.
4. Caller A dials ***87** then dials caller C's number, and then **#** (or wait for 4 seconds)
5. Caller A will hear the confirm tone. Then, A can hang up.

NOTE: "*Enable Call Feature*" must be set to "Yes" in web configuration page.
Caller A can place a call on hold and wait for one of three situations:

1. A quick confirmation tone (similar to call waiting tone) followed by a dial-tone. This indicates the transfer is successful (transferee has received a 200 OK from transfer target). At this point, Caller A can either hang up or make another call.
2. A quick busy tone followed by a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
3. Continuous busy tone. The phone has timed out.

Note: continuous busy tone does not indicate the transfer has been successful, nor does it indicate the transfer has failed. It often means there was a failure to receive second NOTIFY – check firmware for most recent release.

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Attended Transfer

Assume that Caller A and B are in conversation. Caller A wants to *Attend Transfer* B to C:

1. Caller A presses **FLASH** on the analog phone for dial tone.
2. Caller A then dials Caller C's number followed by # (or wait for 4 seconds).
3. If Caller C answers the call, Caller A and Caller C are in conversation. Then A can hang up to complete transfer.
4. If Caller C does not answer the call, Caller A can press "flash" to resume call with Caller B.

NOTE: When Attended Transfer fails and A hangs up, the HT70X will ring back user A to remind A that B is still on the call. A can pick up the phone to resume conversation with B.

3-Way Conferencing

The HT701/702/704 supports Bellcore style 3-way Conference.

Instructions for 3-way conference:

Assume that call party A and B are in conversation. Caller A(HT70X) wants to bring third Caller C into conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials C's number then # (or wait for 4 seconds).
3. If C answers the call, then A presses FLASH to bring B, C in the conference.
4. If C does not answer the call, A can press FLASH back to talk to B.
5. If A presses FLASH during conference, C will be dropped out.
6. If A hangs up, the conference will be terminated for all three parties when configuration "Transfer on Conference Hang up" is set to "No". If the configuration is set to "Yes", A will transfer B to C so that B and C can continue the conversation.

Fax Support

HT70X supports FAX in two modes: 1) T.38 (Fax over IP) and 2) fax pass through. T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting Fax mode to be T.38 (default). If the service provider does not support T.38, pass-through mode can be used.

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CALL FEATURES

The HT70X supports all the traditional and advanced telephony features.

Table 7: HT70X Call Features

Key	Call Features
***02**	**Forcing a Codec** (per call) *027110 (PCMU), *027111 (PCMA), *02723 (G723), *02729 (G729), *0272616 (G726-r16), *0272624 (G724-r24), *0272632 (G726-r32), *0272640 (G726-r40), *027201 (iLBC)
***03**	**Disable LEC** (per call) Dial "*03" + " number ". **No** dial tone is played in the middle.
***16**	**Enable SRTP**
***17**	**Disable SRTP**
***30**	**Block Caller ID** (for all subsequent calls)
***31**	**Send Caller ID** (for all subsequent calls)
***47**	**Direct IP Calling.** Dial "*47" + "IP address". **No** dial tone is played in the middle. Detail see Direct IP Calling section on page 12.
***50**	**Disable Call Waiting** (for all subsequent calls)
***51**	**Enable Call Waiting** (for all subsequent calls)
***67**	**Block Caller ID** (per call). Dial "*67" + " number ". **No** dial tone is played in the middle.
***82**	**Send Caller ID** (per call). Dial "*82" + " number ". **No** dial tone is played in the middle.
***69**	**Call Return Service:** Dial *69 and the phone will dial the last incoming phone number received.
***70**	**Disable Call Waiting** (per call). Dial "*70" + " number ". **No** dial tone is played in the middle.
***71**	**Enable Call Waiting** (per call). Dial "*71" + " number ". **No** dial tone is played in the middle.
***72**	**Unconditional Call Forward:** Dial "*72" and then the forwarding number followed by "**#**". Wait for dial tone and hang up. (dial tone indicates successful forward)
***73**	**Cancel Unconditional Call Forward.** To cancel "Unconditional Call Forward", dial "*73", wait for dial tone, then hang up.
***74**	**Enable Paging Call:** Dial "*74" and then the destination phone number you want to page.
***78**	**Enable Do Not Disturb (DND):** When enabled all incoming calls are rejected.
***79**	**Disable Do Not Disturb (DND):** When disabled, incoming calls are accepted.
***87**	**Blind Transfer**
***90**	**Busy Call Forward:** Dial "*90" and then the forwarding number followed by "**#**". Wait for dial tone then hang up.
***91**	**Cancel Busy Call Forward.** To cancel "Busy Call Forward", dial "*91", wait for dial tone, then hang up.
***92**	**Delayed Call Forward.** Dial "*92" and then the forwarding number followed by "**#**". Wait for dial tone then hang up.
***93**	**Cancel Delayed Call Forward.** To cancel Delayed Call Forward, dial "*93", wait for dial tone, then hang up.
Flash/Hook	Toggles between active call and incoming call (call waiting tone). If not in conversation, flash/hook will switch to a new channel for a new call.
#	Pressing pound sign will serve as **Re-Dial** key.

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CONFIGURATION GUIDE

Configuring the HT70X through Voice Prompts

DHCP MODE

Select voice menu option 01 to enable HT70X to use DHCP.

STATIC IP MODE

Select voice menu option 01 to enable HT70X to use STATIC IP mode, then use option 02, 03, 04, 05 to set up IP address, Subnet Mask, Gateway and DNS server respectively.

FIRMWARE SERVER IP ADDRESS

Select voice menu option 13 to configure the IP address of the firmware server.

CONFIGURATION SERVER IP ADDRESS

Select voice menu option 14 to configure the IP address of the configuration server.

UPGRADE PROTOCOL

Select voice menu option 15 to choose firmware and configuration upgrade protocol. User can choose between TFTP and HTTP.

FIRMWARE UPGRADE MODE

Select voice menu option 17 to choose firmware upgrade mode among the following three options:

1) Always check, 2) check when pre/suffix changes, and 3) never upgrade.

Configuring the HT70X Via Web Browser

HT70X has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow users to configure the HT70X through a web browser such as Microsoft's IE, AOL's Netscape or Mozilla Firefox installed on Windows or Unix OS. (Macintosh OS is not included).

Access the Web Configuration Menu

1. Find the IP address of the HT70X using voice prompt menu option 02.
2. Open a web browser, type the IP address. You will see the log in page of the device.

Note:

- The IVR announces 12 digits IP address, you need to strip out the leading "0" in the IP address. For ex. IP address: 192.168.001.014, you need to type in http://192.168.1.14 in the web browser.

Once the HTTP request is entered and sent from a web browser, the user will see a log-in screen. There are two default passwords for the login page:

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User Level:	Password:	Web pages allowed:
End User Level	123	Only Status and Basic Settings
Administrator Level	admin	Browse all pages

The password is case sensitive with maximum length of 25 characters. The factory default password for End User and administrator is “123” and “admin” respectively. Only an administrator can access the “ADVANCED SETTING”, “FXS PORTs” configuration pages. Please reference the GUI pages using the following link: http://www.grandstream.com/products/ht_series/HT70x/documents/HT70x_gui.zip .

NOTE: If you **cannot** log into the configuration page by using the default password, please check with the VoIP service provider. It is most likely the VoIP service provider has provisioned the device and configured for you therefore the password has already been changed.

Important Settings

The end-user must configure the following settings according to the local environment.
NOTE: Most settings on the web configuration pages are set to the default values.

NAT Settings

If you plan to keep the Handy Tone within a *private network* behind a firewall, we recommend using STUN Server. The following three (3) settings are useful in the STUN Server scenario:

1. **STUN Server** (under Advanced Settings webpage)
   Enter a STUN Server IP (or FQDN) that you may have, or look up a free public STUN Server on the internet and enter it on this field. If using Public IP, keep this field blank.
2. **Use Random Ports** (under Advanced Settings webpage)
   This setting depends on your network settings. Generally if you have multiple IP devices under the same network, it should be set to Yes. If using a Public IP address, set this parameter to No.
3. **NAT Traversal** (under the FXS/Profile web page)
   Set this to Yes when gateway is behind firewall on a private network.

DTMF Methods

DTMF Settings are in FXS portX/ProfileX page.

- DTMF in-audio
- DTMF via RTP (RFC2833)
- DTMF via SIP INFO

Set priority of DTMF methods according to your preference. This setting should be based on your server DTMF setting.

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PREFERRED VOCODER (CODEC)

The HT70X supports a broad range of voice codecs. Under Profile web pages, choose your preferred order of different codecs:

- PCMU/A (or G711μ/a)
- G729 A/B/E
- G723.1
- G726 (16/24/32/40)
- iLBC

Table 8: Basic Settings

End User Password	Password to access the Web Configuration Menu. This field is case sensitive with a maximum length of 25 characters.
Web Port	By default, HTTP uses port 80. This field is for customizable web port.
Telnet Server	Default is set to YES.
IP Address	There are two modes to operate the HT70X: **DHCP mode**: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The HT701 acquires its IP address from the first DHCP server it discovers from the LAN it is connected. *Using the PPPoE feature*: set the PPPoE account settings. The HT70x will establish a PPPoE session if any of the PPPoE fields is set. **Static IP mode:** configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields. These fields are set to zero by default.
DHCP hostname	Default is **blank**. This option specifies the name of the client. This field is optional but may be required by some Internet Service Providers.
DHCP domain	Default is **blank**. This option specifies the domain name that client should use when resolving hostnames via the Domain Name System.
DHCP vendor class ID	Default is **HT7XX**. Used by clients and servers to exchange vendor-specific information..
PPPoE account ID	PPPoE username. Necessary if ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.
PPPoE password	PPPoE account password.
PPPoE Service Name	Default is **blank**. This field is optional. If your ISP uses a service name for the PPPoE connection, enter the service name here.
Preferred DNS server	The preferred DNS Server to be used
Time Zone	Controls how the date/time is displayed according to the specified time zone.

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Self-Defined Time Zone	The syntax is **std offset dst [offset],start[/time],end[/time]** Default is set to : **MTZ+6MDT+5,M3.2.0,M11.1.0** **MTZ+6MDT+5,** Time zone with 6 hours offset with 1 hour ahead which is the US central time. It is positive (+) if the local time zone is west of the Prime Meridian and negative (-) if it is east. **Prime Meridian (a.k.a: International or Greenwich Meridian)** **M3.2.0,M11.1.0** The 1st number indicates Month: 1,2,3,..,12 (for Jan, Feb, .., Dec) The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3rd Tuesday etc) The 3rd number indicates Weekday: 0,1, 2, ..,6(for Sun, Mon, Tue, .., Sat) Therefore, this example is the DST which starts from the second Sunday of March to the 1st Sunday of November.
Allow DHCP server to set Time Zone	Default No. Let the DHCP server handle the Time Zone
Language	Languages supported with voice prompt and web interface, except Spanish that it is only in IVR.
Reset Type	Gives the user the option to set to default all VoIP related configuration (mainly everything located on FXS/Profile page), all ISP (Internet Service Provider) configuration which may affect the IP address, or both at the same time. Note: After you choose the reset type, you will have to click the reset button for it to take effect.

In addition to the Basic Settings configuration page, end users also have access to the Device Status page.

Table 9: Status Page

MAC Address	The device ID, in HEX format. This is very important ID for ISP troubleshooting. The Mac address will appear in this place. The MAC address will be used for provisioning and can be found on the label coming with original box and on the label located on the bottom panel of the device.
IP Address	This field shows the IP address of the HT70X.
Product Model	This field contains the product model info.
Hardware Version	This field shows the hardware revision of the unit and the part number.
Software Version	***Program:*** This is the main software release. *This number is always used for firmware upgrade.* Current release is 1.0.0.18 Boot and Loader are seldom changed. ***Bootloader:*** current version is 1.0.0.6. ***Core:*** current version 1.0.0.18 ***Base:*** current version is 1.0.0.18
System Up Time	Shows system up time since the last reboot.
PPPoE Link Up	Indicates whether the PPPoE connection is up if the HT70X is connected to DSL modem.
NAT	This filed indicates the type of NAT connection used by the HT70X.

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Port Status	Displays relevant information regarding the FXS port. **Port** | **Hook** | **Registration** | **DND** | **Forward** | **Busy Forward** | **Delayed Forward** **FXS** | **On Hook** | **Registered** | Yes | 613 | | • FXS port is registered with SIP Server. • FXS Port user has set Do Not Disturb. • FXS Port user has set his calls to be unconditionally forwarded to ext 613. • FXS Port user has not set Busy or Delay call Forward.

Advanced User Configuration

Log in to the advanced user configuration page the same way as for the basic configuration page. The password is case sensitive and the factory default password for Advanced User is "*admin*".

Advanced User configuration includes the end user configuration and the advanced configurations including: a) SIP configuration, b) Codec selection, c) NAT Traversal Setting and d) other miscellaneous configuration. HT701 and HT702 FXS SIP account(s) have its own configuration page. HT704 has two (2) profiles for the four (4) FXS ports.

Table 10: Advanced Settings

Admin Password	This contains the password to access the Advanced Web Configuration page. This field is case sensitive. Only the administrator can configure the "Advanced Settings" page. Password field is purposely left blank for security reasons after clicking update and saved. The maximum password length is 25 characters.
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is **48**.
Layer 2 QoS	Value used for layer 2 VLAN tag. Default setting is **blank**.
STUN Server	IP address or Domain name of the STUN server.
Keep-alive interval	This parameter specifies how often the HT70X sends a blank UDP packet to the SIP server in order to keep the "hole" on the NAT open. Default is 20 seconds. Minimum value is 20 seconds.
Use STUN to detect network connectivity	Use STUN keep-alive to detect WAN side network problems. If keep-alive request does not yield any response for configured number of times, the device will restart the TCP/IP stack. If the STUN server does not respond when the device boots up, the feature is disabled. Default setting is **No.**
Firmware Upgrade and Provisioning	Enables HT70X to download firmware or configuration file through either the TFTP, HTTP or HTTPS server.

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Via TFTP Server	This is the IP address of the configured TFTP server. If selected and it is non-zero or not blank, the HT70X retrieves the new configuration file or new code image from the specified TFTP server at boot time. After 5 attempts, the system will timeout and will start the boot process using the existing code image in the Flash memory. If a TFTP server is configured and a new code image is retrieved, the new downloaded image is saved into the Flash memory. Note: Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. Depending on the local network, this process can take up to 15 or 20 minutes.
Via HTTP / HTTPS Server	The URL for the HTTP/HTTPS server used for firmware upgrade and configuration via HTTP. *For example*, http://provisioning.mycompany.com:6688/Grandstream/1.0.0.67"**:6688**" is the specific TCP port where the HTTP or HTTPS server is listening; it can be omitted if using default port 80. Note: If Auto Upgrade is set to No, HT70X will only do HTTP/HTTPS download once at boot up.
Firmware Server Path	IP address or domain name of firmware server.
Config Server Path	IP address or domain name of configuration server.
XML Config File Password	The password used for encrypting the XML configuration file using OpenSSL. This is required for the phone to decrypt the encrypted XML configuration file.
HTTP/HTTPS User Name	The user name needed to authenticate with the HTTP/HTTPS server.
HTTP/HTTPS Password	The password needed to authenticate with the HTTP/HTTPS server.
Firmware File Prefix	Default is **blank**. If configured, HT701 will request firmware file with the prefix. This setting is useful for ITSPs. End user should keep it blank.
Firmware File Postfix	Default is **blank**. End user should keep it blank.
Config File Prefix	Default is **blank**. End user should keep it blank.
Config File Postfix	Default is **blank**. End user should keep it blank.
Allow DHCP Option 66 to override server	If set to "Yes", configuration and upgrade server information can be obtained using DHCP option 66 from DHCP server located in customer's environment. Default setting is **Yes.**
Automatic Upgrade	Choose "*Yes"* to enable automatic upgrade and provisioning. If select "Check every minutes" input the amount of minutes you want it to check for update. If select "Yes, daily at hour" make sure to input the hour of the day you want it to check for update, e.g. for 11 pm type 23. If select "Yes, weekly on day" make sure you input the day of the week (in format 0-6, 0 is Sunday) you want it to check for update. When set to No, HT70X will only do the following option you select; "Always check for New Firmware at Boot up" will check for new firmware every time the device reboots. "Check New Firmware only when F/W pre/suffix changes" will check for updates only when the pre/suffix has been changed.
Authenticate Conf File	If set to Yes, config file is authenticated before acceptance. This protects the configuration from an unauthorized change.
Firmware Key	Used for firmware encryption. Should be 32 digit in hexadecimal representation. End user should keep it blank.
SSL Certificate	The user specify SSL certificate used for SIP over TLS in X.509 format.

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SSL Private Key	The user specify SSL private key used for SIP over TLS in X.509 format.
SSL Private Key Password	User specify password to protect the private key above.
ACS URL	User specify the Auto Configuration Server's URL (TR-069 protocol)
ACS Username	User specify the ACS Username
ACS Password	User specify the ACS password
Periodic Inform Enable	Default is **No**. If set to YES, device will send inform packets to the ACS
Periodic Inform Interval	Frequency that the inform packets will be sent out to the ACS
Connection Request Username	Set a user name for the ACS to connect to this device
Connection Request Password	Set a password for the ACS to connect to this device
System Ring Cadence	Configuration option is set ring cadence on all FXS ports for all incoming calls. (Syntax: c=on1/off1-on2/off2-on3/off3(only 3 cadences maximum)) Default is set to **c=2000/4000;** (US standards)
Call Progress Tones	Using these settings, users can configure tone frequencies and cadence according to their preference. By default they are set to North American frequencies. Configure these settings with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous tone, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Example configuration for N.A. Dialtone: **f1=350@-13,f2=440@-13,c=0/0;** **Syntax: f1=freq@vol, f2=freq@vol, c=on1/off1-on2/off2-on3/off3; [...]** **(Note: freq: 0 - 4000Hz; vol: -30 - 0dBm)**
Lock Keypad Update	Default is **No**. If set to "Yes", the configuration update via keypad is disabled.
Disable Voice Prompt	Default is **No**. Disables the voice prompt configuration.
Disable Direct IP Call	Default is **No**. Disables the Direct IP Call function.
Failover to FXO Gateway	Default is **Disable.** If Enabled, the unit will start routing the calls to the FXO Gateway configured via Direct IP Call When it loses registration from your SIP Server. **HT704 Only** The use of this option presumes a configured GXW410x or another FXO gateway with an active PSTN line connection.
FXO Gateway	IP or Domain Name of the FXO Gateway that will be used as Failover. **HT704 Only**
NTP server	URI or IP address of the NTP (Network Time Protocol) server. This parameter synchronizes the date and time.
Allow DHCP option 42 to override NTP serve	Default **NO.** Enables the DHCP server to handle the NTP Server via Option 42
Syslog Server	The IP address or URL of System log server. This feature is especially useful for the ITSP (Internet Telephone Service Provider)

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Syslog Level	Select the HT701 to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message *Example*: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
Send SIP Log	Default is **No.** If Enabled, the HT70X will send a replicate of the SIP packets on the syslog.
Primary RADIUS Server	Set the IP or FQDN of the Primary RADIUS Server. **HT704 Only** Remote Authentication Dial In User Service (RADIUS). HT704 supports RADIUS for authentication, authorization and billing purposes. Primary and secondary RADIUS server configurations are available to provide redundancy to this feature. In case Primary Radius server becomes unusable, RADIUS requests will be automatically sent to the secondary server. When at least one RADUIS server was configured, the device will allow users to make phone calls only after authorization from RADIUS server has been received. CDR (Call Detail Record) is also sent to the RADIUS server for billing purposes. RAIDUS server can send requests to terminate calls when run out of pre-paid credit.
Primary RADIUS Auth Port	Default is **1812.** Specifies the port to be used for the Primary RADIUS Authentication. **HT704 Only**
Primari RADIUS Acct Port	Default is **1813.** Specifies the port to be used for the Primary RADIUS Account. **HT704 Only**
Primary RADIUS Server Secret	Specifies the secret string to be used to authenticate the RADIUS connection to the Primary Server. It should match RADIUS configuration. **HT704 Only**
Secondary RADIUS Server	Set the IP or FQDN of the Secondary RADIUS Server. **HT704 Only** In case Primary Radius server becomes unusable, secondary will take role of primary and will manage credit recourses in the network
Secondary RADIUS Auth Port	Default is **1812.** Specifies the port to be used for the Secondary RADIUS Authentication. **HT704 Only**
Secondary RADIUS Acct Port	Default is **1813.** Specifies the port to be used for the Secondary RADIUS Account. **HT704 Only**
Secondary RADIUS Server Secret	Specifies the secret string to be used to authenticate the RADIUS connection to the Secondary Server. It should match RADIUS configuration **HT704 Only**

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RADIUS Timeout	Default is **2. HT704 Only**
RADIUS Retry	Default is **3. HT704 Only**
Download Device Configuration	Allows user to download and save a text file containing all the P values of each setting as configured at that point on the unit. (Note: For Security Reasons, all Passwords won't be Downloaded)
Upload Firmware	Allows the user to upgrade the firmware with a single firmware file by browsing and loading the file from your computer (local directory).

Table 11: Account Settings

Profile Active	When set to Yes the FXS port or Profile (for HT704) is activated.
Primary SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Failover SIP Server	Failover SIP Server's IP address or Domain name in case primary server does not respond.
Prefer Primary SIP Server	Default is **No**. If set to yes, it will register to Primary Server if registration with Failover server expires
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by HT70X for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can correct the problem.
SIP transport	User can select UDP or TCP or TLS. Default is **UDP.**
NAT Traversal (STUN)	This parameter defines whether or not the HT70X NAT traversal mechanism is activated. If activated (by choosing "Yes") and a STUN server is also specified, then the HT70X performs according to the STUN client specification. Using this mode, the embedded STUN client will detect if and what type of firewall/NAT. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the HT70X will use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" *with no specified STUN server*, the HT70X will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number. **HT701 and HT702 only**
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID. **HT701 and HT702 only**
Authenticate Password	SIP service subscriber's account password. **HT701 and HT702 only**
Name	SIP service subscriber's name for Caller ID display. **HT701 and HT702 only**
DNS Mode	One from the 3 modes are available for "DNS Mode" configuration: -A Record (for resolving IP Address of target according to domain name) -SRV (DNS SRV resource records indicates how to find services for various protocols) -NAPTR/SRV (Naming Authority Pointer according to RFC 2915) One mode can be chosen for the client to look up server. The default value is "A Record"
Tel URI	The default setting is **"Disabled".** If the phone has an assigned PSTN Number, this field should be set to "User=Phone" then a "User=Phone" parameter will be attached to the "From header" in the SIP

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	request to indicate the E.164 number. If server supports TEL URI format, then this option needs to be selected.
SIP Registration	Controls whether the HT701 needs to send REGISTER messages to the proxy server. The default setting is **Yes**.
Unregister on Reboot	Default is **No**. If set to Yes, the SIP user's registration information will be cleared on reboot.
Outgoing Call without Registration	Default is **No**. If set to "Yes," user can place outgoing calls even when not registered (if allowed by Internet Telephone Service Provider) but is unable to receive incoming calls.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the HT70X refreshes its registration with the specified registrar. The default interval is **60** minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Registration Retry Wait Time	Retry registration if the process failed. Default is **20** seconds.
Local SIP port	Defines the local SIP port the HT70X will listen and transmit. The default value for FXS port is **5060**.
Local RTP port	Defines the local RTP-RTCP port pair the HT70X will listen and transmit. It is the base RTP port for channel 0. When configured, **channel 0** uses this port _value for RTP and the port_value+1 for its RTCP The default value for FXS port is **5004**.
Use Random Port	Default is **No**. This parameter forces the random generation of both the local SIP and RTP ports when set to Yes. This is usually necessary when multiple HT70X are behind the same NAT.
Refer to Use Target Contact	Default is **No**. If set to YES, then for Attended Transfer, the "Refer-To" header uses the transferred target's Contact header information.
Transfer on Conference Hang up	Default is **No**. In which case if the conference originator hangs up the conference will be terminated. When option YES is chosen, originator will transfer other parties to each other so that B and C can choose to either continue the conversation or hang up.
Enable Ring-Transfer	Default is **No**, this will create a Semi-Attendant Transfer. When set to Yes, device can transfer the call upon receiving ring back tone or SIP message 180.
Disable Bellcore Style 3-Way Conference	Default is **No**. you can make a Conference by pressing 'Flash' key. If set to **Yes**, you need to dial *23 + second callee number.
Remove OBP from Route Header	Default is **No**. When option YES is chosen, the Out Bound Proxy will be removed from Route header.
Support SIP Instance ID	Default is **Yes**. If set to Yes, the contact header in REGISTER request will contain SIP Instance ID as defined in IETF SIP Outbound draft.
Validate incoming SIP message	Default is **No**. If set to yes all incoming SIP messages will be strictly validated according to RFC rules. If message will not pass validation process, call will be rejected.
Check SIP User ID for incoming INVITE	Default is **No.** Check the incoming SIP User ID in Request URI. If they don't match, the call will be rejected. If this option is enabled, the device will not be able to make direct IP calls.
Allow Incoming SIP Messages from SIP Proxy Only	Default is **No.** Check the incoming SIP messages. If they don't come from the SIP proxy, they will be rejected. If this option is enabled, the device will not be able to make direct IP calls.
SIP T1 Timeout	T1 is an estimate of the round-trip time between the client and server transactions. If the network latency is high, select larger value for more reliable usage. Default is **0.5 Sec.**
SIP T2 Interval	Maximum retransmission interval for non-INVITE requests and INVITE responses. Default is **4 Sec.**
DTMF Payload Type	Sets the payload type for DTMF using RFC2833. Default is **101.**

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Preferred DTMF method	The HT70X supports up to 3 different DTMF methods including in-audio, via RTP (RFC2833) and via Sip Info using SIP INFO messages. The user can configure DTMF method in a priority list.
Disable DTMF Negotiation	Default is **No**. If set to yes, use above DTMF order without negotiation
Send Flash Event	Default is **No.** If set to yes, flash will be sent as DTMF event.
Enable Call Features	Default is **Yes**. (If Yes, call features using star codes will be supported locally)
Offhook Auto-Dial	This parameter allows users to configure a User ID or extension number that is automatically dialed when off-hook. Only the user part of a SIP address needs to be entered here. The HT70X will automatically append the "@" and the host portion of the corresponding SIP address. **HT701 and HT702 only**
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
Distinctive Ring Tone	Custom Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY uses this ring tone when the incoming call is from the Caller ID. System Ring Tone is used for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for **all** incoming calls using the FXS port or Profile. Distinctive ring tones can be configured not only for matching a whole number, but also for matching prefixes. In this case symbol * (star) will be used. **For example:** if configured as *617, Ring Tone 1 will be used in case of call arrived from the area code 617. Any other incoming call will ring using cadence defined in parameter System Ring Cadence located under Advanced Settings Configuration page. **Note:** If server supports Alert-Info header and standard ring tone set (Bellcore) or distinctive ring tone 1-10 is specified, then the ring tone in the Alert-Info header from server will be used. Bellcore rings and tones are independent from custom ring tones. The custom ring tones can also be specified by alert-info header, for example Alert-Info: <http://127.0.0.1>;info=ring5
Disable Call Waiting	Default is **No**. If set to YES Call Waiting indication information will not be provided to analog phone connected to this FXS port.
Disable Call-Waiting Caller ID	Default is **No**. If set to YES Call Waiting caller ID will not be provided to analog phone connected to this FXS port.
Disable Call Waiting Tone	Default is **No**. This is to disable the stutter Call Waiting Tone when a Call Waiting information arrives. The CWCID information will still be displayed.
Disable Receiver Offhook Tone	Default is **No.** If set to yes, it will disable the warning to alert that the phone has been left off-hook for an extended period of time.
Disable Reminder Ring for On-Hold Call	Default is **No**. Do not play the reminder ring when this is set to Yes.
Disable Visual MWI	Default is **No.** If set to "Yes", the MWI information will not be transferred to the analog phone connected to the FXS port.
Ring Timeout	Default value is **60 Sec.** Incoming call will stop ringing when not picked up given a specific period of time.
Hunting Group Ring Timeout	Default value is **20 Sec.** If call is not answered within this designated time period, the callwill be forwarded to the next member of a Hunt Group. **HT704 only**
Hunting Group Type	Linear and Circular. Default is **Circular.** Linear style will sort the call to the lowest-numbered available line, this is also called "serial hunting". Circular style will distribute the calls "round-robin". If a call is assigned to line 1, the next call goes to 2 and the next to 3. The succession throughout each of the lines continues even if one of the previous lines becomes available. When the end of the hunt group is reached, the hunting starts over at the firstline. Lines are skipped if they are still busy on a previous call. **HT704 only**
Delayed Call Forward Wait Time	Default value is **20 seconds**. In case this feature activated using * codes (*92 code), the call will be forwarded after this preconfigured amount of time.

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No Key Entry Timeout	Default is **4 seconds**. Dialing process is completed and outgoing call is initiated if no key entry occurs during this preconfigured interval.
Early Dial	Default is **No**. Use only if proxy supports 484 response. This parameter controls whether the phone will send an early INVITE each time a key is pressed when a user dials a number. If set to "Yes", an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the "(Re-)Dial" button is pressed or after about 5 seconds have elapsed if the user forgets to press the "Re-Dial" button. The "Yes" option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). *This feature does NOT work with and should NOT be enabled for direct IP-to-IP calling*.
Dial Plan Prefix	Sets the prefix added to each dialed number.
Use # as Dial Key	Default is **Yes.** It allows users to configure the "**#**" key as the "Send" (or "Dial") key. If set to "Yes", "**#**" will send the number. In this case, this key is essentially equivalent to the "Dial" key. If set to "No", this "**#**" key can be included as part of number.

Dial Plan	**Dial Plan Rules**: 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d 2. Grammar: x - any digit from 0-9; • xx+ - at least 2 digits number; • xx. – at least 2 digit number. • ^ - exclude; • [3-5] - any digit of 3, 4, or 5; • [147] - any digit 1, 4, or 7; • <2=011> - replace digit 2 with 011 when dialing • < =1> - add a leading 1 to all numbers dialed, vice versa will remove a 1 from the number dialed • | - or • **Example 1:** {[369]11 | 1617xxxxxxx} – Allow 311, 611, 911, and any 11 digit numbers with leading digits 1617 **Example 2:** {^1900x+ | <=1617>xxxxxxx} – Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers • **Example 3:** {1xxx[2-9]xxxxxx | <2=011>x+} –Allow any combinations of numbers with 11 digits which has a leading digit 1, but 5th digit cannot be 0 or 1. Or any length of numbers with a minimum of 2 digits beginning with 2, with the leading digit replaced with 011. 3. Default: Outgoing - {x+} Example of a simple dial plan used in a Home/Office in the US: **{ ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 }** ***Explanation*** of example rule (reading from left to right): • **^1900x. -** prevents dialing any number started with 1900 • **<=1617>[2-9]xxxxxx -** allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically • **1[2-9]xx[2-9]xxxxxx |-** allows dialing to any US/Canada Number with 11 digits length • **011[2-9]x. -** allows international calls starting with 011 • **[3469]11 -** allow dialing special and emergency numbers 311, 411, 611 and 911 **Note:** In some cases user wishes to dial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature and the Dial Plan should be: **{ *x+ }**.

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Subscribe for MWI	Default is **No**. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Send Anonymous	Default is **No**. If this parameter is set to "Yes", the "From" header along with Privacy and P_ Asserted_Identity headers in outgoing INVITE message will be set to anonymous, blocking Caller ID.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Special Feature	Default is **Standard**. Choose the selection to meet some special requirements from Softswitch vendors.
Session Expiration	Grandstream implemented SIP Session Timer. The session timer extension enables SIP sessions to be periodically "refreshed" via a SIP request (UPDATE, or re-INVITE. Once the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is **180 seconds**.
Min-SE	The minimum session expiration (in seconds). The default value is **90 seconds**.
Caller Request Timer	Default is **No**. If selecting "Yes" the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	Default is **No**. If selecting "Yes" the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	Default is **No**. If selecting "Yes" the phone will use session timer even if the remote party does not support this feature. Selecting "No" will allow the phone to enable session timer only when the remote party support this feature. To turn off Session Timer, select "No" for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	Default is **Omit.** As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	Default is **UAC.** As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select "Yes" to use INVITE method to refresh the session timer. Default is **No**,
Send Re-INVITE After Fax	Default is **No**, If set to "Yes", device will send an INVITE with audio vocoders upon completion of Fax to continue session in audio only.
Enable Silence Detection for Fax Disconnect	For fax machines that do not send a Disconnect when fax is done. This option Enables/Disables the detection of silence in order to know the fax has finished. The silence period is non-configurable and fixed to 7 seconds. Default is **No**,
Enable 100rel	Default is **No**, If set to Yes, Enables the use of PRACK (Provisional Acknowledgment) method.
Use First Matching Vocoder in 200OK SDP	Default is **No**. If set to "Yes", device will include only the first match vocoder in its 200OK response, otherwise it will include all match vocoders in same order received in INVITE.
Preferred Vocoder	The HT70X supports up to 5 different Vocoder types including G.711 A-/U-law, G.726-32, G.723.1, G.729A/B/E, iLBC. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder is entered by choosing the appropriate option in "Choice 1". The last Vocoder is entered by choosing the appropriate option in "Choice 8". Vocoder types can also be changed per call basis by using a star code. Please see the "Call features" section.
G723 Rate	Default is **6.3kbps**. Defines the encoding rate for G.723.1 vocoder.
iLBC Frame Size	Default is **20ms**. Sets the iLBC frame size in 20ms or 30ms.
iLBC Payload type	Defines payload type for iLBC. Default value is **97**. The valid range is between 96 and 127.
VAD	Default is **No**. VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.

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Symmetric RTP	Default is **No**. When set to Yes the device will change the destination to send RTP packets to the source IP address and port of the inbound RTP packet last received by the device.
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Re-Invite after Fax Tone Detection Mode	Default is **Enabled.** It makes the unit send out the re-INVITE for T.38 or Fax Pass Through if a fax tone is detected.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions. Default is **Adaptive.**
Jitter Buffer Length	Select Low, Medium or High based on network conditions. Default is **Medium.** • High (initial 200ms, min 40ms, max 600ms) Note: not all vocoders can meet the high requirement • Medium (initial 100ms, min 20ms, max 200ms) • Low (initial 50ms, min 10ms, max 100ms)
SRTP Mode	This option defines different implementation of support SRTP (squired RTP) transmission mode. Select between Disabled, Enabled but not Forced, and Enabled and Forced. Default is **Disabled.**
SLIC Setting	Dependent on standard phone type (and location)
Caller ID Scheme	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF, SIN 227 – BT, & NTT Japan
Polarity Reversal	Default is **No**. If set to "Yes", polarity will be reversed upon call establishment and termination.
Loop Current Disconnect	Default is **No**. Set it to Yes if the traditional PBX you are using with HT70X uses this method for signaling call termination. Method initiates short voltage drop on the line when remote (VoIP) side disconnects an active call.

Loop Current Disconnect Duration	Default value is **200.** Here can be configured duration of such voltage drop described in topic above. HT70X supports a Duration Range from 100 to 10000 ms.
Hook Flash Timing	Time period when the cradle is pressed (Hook Flash) to simulate FLASH. To prevent unwanted activation of the Flash/Hold and automatic phone ring-back, adjust this time value. Default values are **300 minimum and 1100 maximum.** HT70X supports a range from 40 to 2000 ms.
On Hook Timing	On-hook timing is the minimum time for an on-hook event to be validated. Default value is **400** . HT70X supports a range from 40 to 2000 ms.
Gain	Voice path volume adjustment. • Rx is a gain level for signals transmitted by FXS • Tx is a gain level for signals received by FXS. Default = **0dB** for both parameters. Loudest volume: +6dB Lowest volume: -6dB. User can adjust volume of call on either end using the Rx Gain Level parameter and the Tx Gain Level parameter located on the FXS Port Configuration page. If call volume is too low when using the FXS port (ie. the ATA is at user site), adjust volume using the Rx Gain Level parameter under the FXS Port Configuration page. If voice volume is too low at the other end, user may increase the far end volume using the Tx Gain Level parameter under the FXS Port Configuration page.
Disable Line Echo Canceller (LEC)	Default is **No.** If set to "Yes" LEC will be disabled per call base. Recommended for FAX/Data calls.
Ring Tones	This function lets you configure ring tone cadence preferences. User has 10 choices. The configuration, completed in Distinctive Ring Tones block in the same page, applies to ring tones cadences configured here.

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*Table 12: **HT704** FXS Ports Settings*

SIP Use ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
Password	SIP service subscriber's account password for HT704 to register to (SIP) servers of ITSP.
Name	Any name to identify this specific user.
Profile ID	Select the corresponding Profile ID between Profile 1 and Profile 2.
Hunting Group	This feature enables the HT704 to register all existing FXS ports with the same phone number. Each incoming call will be routed to first available port in Linear or Circular mode. User may configure all ports as members of the same Hunting Group or it may configure different port combinations for more than one Hunting Group. For example: Ports 1, 3 and 5 are members of the same Hunting Group, the rest of the ports may have separate numbers and may be reached independently. Any port, member of a Hunting Group that is not registered with a SIP account, will be able to place outbound calls using the SIP credentials of the primary Hunting Group port. For example: Port 1, 2, and 3 are members of the same Hunting Group. Port 1 is registered with a SIP account. Ports 2, and 3 are not registered. Ports 2 and 3 will be able to place outbound calls using the SIP account of port 1. Select appropriate value for Hunting Group feature. The original SIP account should be set to Active while the group members should be set to the port number of the Active Port. Example configuration of a Hunting group: FXS Port #1: SIP UserID and Authenticate ID entered, Hunting group set to "Active" FXS Port #2: SIP UserID and Authenticate ID left blank, Hunting Group set to "1" FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "1" FXS Port #4: SIP UserID and Authenticate ID entered, Hunting group set to "None" Hunting Group 1 contains ports 1, 2, 3. FXS port 4 is registered but it is not added to the Hunting Group 1.
Offhook Auto-dial	This feature allows you to automatically dial the number specified in this field as soon as the port is offhooked, i.e. when the receiver on the phone connected to Port# is picked up.
Map to FXP Port #	This is used only when peering with a Grandstream GXW410x. Default is 1, Supported values are 1-8, meaning line 1 to line 8 of the GXW410x device where the port will be mapped to.
Map to FXO Gateway IP	This is used when peering with an FXO gateway of any brand. You have to specifically mention the IP address where the call will be sent to.
and Port	Sip port that will be annexed to the IP address above.

Saving the Configuration Changes

Click the "Update" button in the Configuration page to save the changes to the HT70X configuration. The following screen confirms that the changes are saved. Reboot or power cycle the HT70X to make the changes take effect.

Rebooting the HT70X from Remote

Press the "Reboot" button at the bottom of the configuration menu to reboot the ATA remotely. The web browser will then display a message window to confirm that reboot is underway. Wait 30 seconds to log in again.

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Configuration through a Central Server

Grandstream HT70X can be automatically configured from a central provisioning system.

When HT70X boots up, it will send TFTP or HTTP/HTTPS request to download configuration file, “cfg000b82xxxxxx” or “cfg00082xxxxxx.xml”, where “000b82xxxxxx” is the MAC address of the HT70X. It will first request “cfg000b82xxxxxx” then “cfg000b82xxxxxx.xml”

The configuration file can be downloaded via TFTP or HTTP/HTTPS from the central server. A service provider or an enterprise with large deployment of HT70X can easily manage the configuration and service provisioning of individual devices remotely from a central server.

Grandstream has a provisioning system called GAPS (Grandstream Automated Provisioning System) that is used to support automated configuration of Grandstream devices. GAPS uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual Grandstream device for firmware upgrade, remote reboot, etc.

Grandstream provides GAPS service to VoIP service providers. Use GAPS for either simple redirection or with certain special provisioning settings. At boot-up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer’s TFTP or HTTP/HTTPS server for further provisioning. Grandstream also provide GAPSLITE software package which contains our NAT friendly TFTP server and a configuration tool to facilitate the task of generating device configuration files.

The configuration tool is now free to end users. The tool and configuration template are available for download from http://www.grandstream.com/support/tools .

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SOFTWARE UPGRADE

Software upgrade can be done via either TFTP or HTTP. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

Firmware Upgrade through TFTP/HTTP/HTTPS

To upgrade via TFTP or HTTP/HTTPS, the "Firmware Upgrade and Provisioning upgrade via" field needs to be set to TFTP HTTP or HTTPS, respectively. "Firmware Server Path" needs to be set to a valid URL of a TFTP or HTTP server, server name can be in either FQDN or IP address format. Here are examples of some valid URL.

e.g. firmware.mycompany.com:6688/Grandstream/1.0.0.18
e.g. 72.172.83.110

NOTES:

- Firmware upgrade server in IP address format can be configured via IVR. Please refer to the CONFIGURATION GUIDE section for instructions. If the server is in FQDN format, it must be set via the web configuration interface.

- Grandstream recommends end-user use the Grandstream HTTP server. Its address can be found at http://www.grandstream.com/support/firmware . Currently the HTTP firmware server IP address is 72.172.83.110. For large companies, we recommend to maintain their own TFTP/ HTTP/HTTPS server for upgrade and provisioning procedures.

- Once a "Firmware Server Path" is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the HT70X will attempt to retrieve the new image files by downloading them into the HT70X's SRAM. During this stage, the HT70X's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP/HTTPS fails for any reason (e.g. TFTP/HTTP/HTTPS server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the HT70X will stop the TFTP/HTTP/HTTPS process and simply boot using the existing code image in the flash.

- Firmware upgrade may take as long as 15 to 30 minutes over Internet, or just 5 minutes if it is performed on a LAN. It is recommended to conduct firmware upgrade in a controlled LAN environment if possible. For users who do not have a local firmware upgrade server, Grandstream provides a NAT-friendly TFTP server on the public Internet for firmware upgrade.

- Grandstream's latest firmware is available at http://www.grandstream.com/support/firmware .

  Oversea users are strongly recommended to download the binary files and upgrade firmware locally in a controlled LAN environment.

Note: Alternatively, the user can upload the firmware single file from his/her computer using a local directory. For more instructions see below. Also the user can download a free TFTP or HTTP server and conduct local firmware upgrade. A free windows version TFTP server is available for download from http://support.solarwinds.net/updates/New-customerFree.cfm. Our latest official release can be downloaded from http://www.grandstream.com/support/firmware .

Instructions for Upload from Local Directory:

1- Download the firmware file from Grandstream web site
2- Unzip it and copy the file in to a folder in your PC

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3- From the HT70X web interface (Advanced Settings page) you can browse your hard drive and select the folder you previously saved the file (ht70xfw.bin)

4- Click “Upload Firmware” and wait few minutes until the new program is loaded.

Note: Always check the status page to see that the program version has changed.

Instructions for local firmware upgrade using TFTP server:

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the HT701 device in the same LAN segment.
3. Please go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Start the TFTP server, in the phone’s web configuration page
5. Configure the Firmware Server Path with the IP address of the PC
6. Update the change and reboot the unit

End users can also choose to download the free HTTP server from http://httpd.apache.org/ or use Microsoft IIS web server.

Configuration File Download

Grandstream SIP Device can be configured via Web Interface as well as via Configuration File through TFTP or HTTP/HTTPS. “Config Server Path” is the TFTP or HTTP/HTTPS server path for configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The “Config Server Path” can be same or different from the “Firmware Server Path”.

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 1 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with “Admin Password” in the ADVANCED SETTINGS page. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When Grandstream Device boots up or reboots, it will issue request for configuration file named “cfgxxxxxxxxxxxx”, where “xxxxxxxxxxxx” is the MAC address of the device, i.e., “cfg000b820102ab”. The configuration file name should be in lower cases.

Firmware and Configuration File Prefix and Postfix

Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. This makes it the possible to store ALL of the firmware with different version in one single directory. Similarly, Config File Prefix and Postfix allows device to download the configuration file with the matching Prefix and Postfix. Thus multiple configuration files for the same device can be stored in one directory.

In addition, when the field “Check New Firmware only when F/W pre/suffix changes” is set to “Yes”, the device will only issue firmware upgrade request if there are changes in the firmware Prefix or Postfix.

Managing Firmware and Configuration File Download

When “Automatic Upgrade” is set “**Yes, every**” the auto check will be done in the minute specified in this field. If set to “**daily at hour (0-23)”**, Service Provider can use P193 (Auto Check Interval) to have the devices do a daily check at the hour set in this field with either Firmware Server or Config Server. If set to “**weekly on day (0-6)”** the auto check will be done in the day specified in this field. This allows the device

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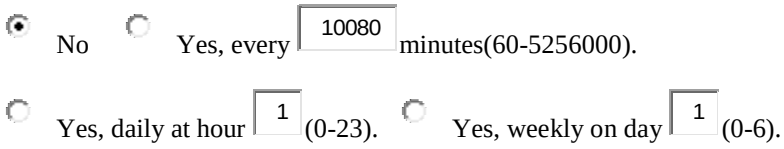
periodically check if there are any new changes need to be taken on a scheduled time. By defining different intervals in P193 for different devices, Server Provider can spread the Firmware or Configuration File download in minutes to reduce the Firmware or Provisioning Server load at any given time.

Automatic Upgrade:



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RESTORE FACTORY DEFAULT SETTING

WARNING! Restoring the Factory Default Setting will DELETE all configuration information of the phone. Please BACKUP or PRINT out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.

FACTORY RESET

There are three (3) methods for resetting your unit:

Reset Button

Reset default factory settings following these four (4) steps:

1. Unplug the Ethernet cable.
2. Locate a needle-sized hole on the back panel of the gateway unit next to the power connection.
3. Insert a pin in this hole, and press for about 7 seconds.
4. Take out the pin. All unit settings are restored to factory settings.

IVR Command

Reset default factory settings using the IVR Prompt (Table 5):

1. Dial "***" for voice prompt.
2. Enter "99" and wait for "reset" voice prompt.
3. Enter the encoded MAC address (Look below on how to encode MAC address).
4. Wait 15 seconds and device will automatically reboot and restore factory settings.

Encode the MAC Address

1. Locate the MAC address of the device. It is the 12 digit HEX number on the bottom of the unit.
2. Key in the MAC address. Use the following mapping:
   - 0-9: 0-9
   - A: 22 (press the "2" key twice, "A" will show on the LCD)
   - B: 222
   - C: 2222
   - D: 33 (press the "3" key twice, "D" will show on the LCD)
   - E: 333
   - F: 3333

For example: if the MAC address is 000***b***8200***e***395, it should be keyed in as "000***222***8200***333***395".

Reset from web interface (Reset Type)

1. From the Advanced Settings Page user can select three types:
   - **Full Reset**: This will make a full reset
   - **ISP Data**: This will reset only the basic settings, like IP mode, PPPoE and Web port
   - **VOIP Data**: This will reset only the data related with a service provider like SIP server, sip user ID, provisioning and others.

NOTE:

1. Factory Reset will be disabled if the "**Lock keypad update**" is set to "Yes".
2. If the HT70X was previously locked by your local service provider, pressing the RESET button will only restart the unit. The device will not return to factory default settings.

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